AMENDMENTS TO THE CLAIMS:

The following claims replace all prior versions and listings of claims in the application:

1. (Currently Amended) A method of ~~communicating~~ speech communication using very low digital data bandwidth, comprising:

providing a bi-directional digital telephony link over a digital packet network between a source terminal and a destination terminal,

wherein the source terminal for a forward link serves as the destination terminal for a reverse link on the digital telephony link;

distinguishing between speech and a pause in the speech communication using an ITU voice activity detection module;

providing a comfort noise simulation of background noise for each distinguished pause in the speech communication in time order with the speech;

translating said speech into text at the source terminal;

communicating said text and said time-ordered simulated comfort noise of each pause in speech across the bi-directional digital telephony link to the destination terminal;

determining a status of the telephony link;

ending the communicating if the telephony link is terminated;

generating a speaker voice profile by training the source terminal to recognize words spoken by a speaker for reproduction of audible speech corresponding to words

spoken by the speaker having audible qualities approximating that of the speaker;

communicating the voice profile across the telephony link from the source terminal to the destination terminal,

wherein the speaker's voice profile contains the information needed to generate the reproduced speech that substantially resembles the sound of the speaker's voice; and

translating said text into reproduced speech received at said destination terminal using the speaker's voice profile <u>and reproducing the simulated background noise in the reproduced speech in time order with each pause in the speech at the source terminal</u> at the destination terminal.

2. (Previously Presented) The method of claim 1, further comprising the step of:

generating said reproduced speech using a default voice profile.

3-6 (Canceled)

7. (Previously Presented) The method of claim 1, further comprising the step of:

generating said reproduced speech using a speaker's voice profile at said destination terminal, wherein said speaker's voice profile contains the information

needed to generate said reproduced speech that substantially resembles the sound of said speaker's voice.

8. (Previously Presented) The method of claim 7, further comprising the steps of:

generating said reproduced speech using a default voice profile of said destination terminal, until said speaker's voice profile has been communicated across said communication link; and

generating said reproduced speech using said speaker's voice profile at said destination terminal, after said speaker's voice profile has been communicated across said communication link.

9. (Previously Presented) The method of claim 1, wherein:

a portion of said training is performed during a portion of the time said speaker is communicating speech across said communication link.

10. (Previously presented) The method of claim 9, wherein:

said speaker's voice profile is periodically updated as said speaker uses said method, and

said updated profile is periodically communicated to said destination terminal.

11-15. (Canceled)

16. (Currently Amended) A method of communicating speech for a telephone conversation across a bi-directional communication link using very low digital data bandwidth, comprising:

providing a bi-directional digital telephony link over a digital network between a source terminal and a destination terminal,

wherein the source terminal for a forward link serves as the destination terminal for a reverse link on the digital telephony link;

<u>distinguishing between a first speaker's speech and a pause in the speech communication of the first speaker using an ITU voice activity detection module;</u>

<u>providing a comfort noise simulation of background noise for each distinguished pause in speech in the speech communication of the first speaker in time order with the first speaker's speech;</u>

translating a first speaker's speech into first text characters at the source terminal;

communicating, from the source terminal, said first text characters <u>and said time-ordered simulated comfort noise of each pause in the first speaker's speech</u> across the digital telephony link to the destination terminal;

determining, from the source terminal, whether the bi-directional digital telephony link has terminated, and if the digital telephony link has terminated then ending the

translating and the communicating from the source terminal;

translating said first text characters into first reproduced speech and reproducing the simulated background noise in the reproduced speech of the first speaker in time order with each pause in the speech at the source terminal at the destination terminal;

distinguishing between a second speaker's speech and a pause in the speech communication of the second speaker using an ITU voice activity detection module;

providing a comfort noise simulation of background noise for each distinguished pause in speech in the speech communication of the second speaker in time order with the second speaker's speech;

translating a second speaker's speech into second text characters at the destination terminal;

communicating said second text characters and said time-ordered simulated comfort noise of each pause in the second speaker's speech across the digital telephony link to the source terminal;

determining, from the destination terminal, whether the bi-directional digital telephony link has terminated, and if the digital telephony link has terminated then ending the translating and the communicating from the destination terminal; and

translating said second text characters into second reproduced speech and reproducing the simulated background noise in the reproduced speech of the second speaker in time order with each pause in the speech at the source terminal at the source terminal.

17. (Previously Presented) The method of claim 16, further comprising:

providing the source terminal with a first voice profile of said first speaker; and

providing the destination terminal with a second voice profile of said second speaker;

communicating said first voice profile across said bi-directional digital telephony link to the destination terminal; and

communicating said second voice profile across said bi-directional digital telephony link to said first terminal;

generating said first reproduced speech using said first speaker's voice profile at the destination terminal;

generating said second reproduced speech using said second speaker's voice profile at the source terminal, wherein

said first speaker's voice profile contains the information needed to generate said first reproduced speech that substantially resembles the sound of said first speaker's voice, and

said second speaker's voice profile contains the information needed to generate said second reproduced speech that substantially resembles the sound of said second speaker's voice.

18. (Previously Presented) The method of claim 17, further comprising:

generating said first reproduced speech using a default voice profile of the

destination terminal, until said first speaker's voice profile has been communicated across the digital telephony link;

generating said second reproduced speech using a default voice profile of the source terminal, until said second speaker's voice profile has been communicated across the digital telephony link;

generating said first reproduced speech using said first speaker's voice profile at said second terminal, after said first speaker's voice profile has been communicated across the digital telephony link; and

generating said second reproduced speech using said second speaker's voice profile at the source terminal, after said second speaker's voice profile has been communicated across the digital telephony link.

19. (Previously Presented) The method of claim 18, wherein:

said first text and said first speaker's voice profile are simultaneously communicated across the digital telephony link; and

said second text and second speaker's voice profile are simultaneously communicated across the digital telephony link.

20. (Previously Presented) The method of claim 18, wherein:

said first speaker's voice profile is provided by first training at the source terminal; and

said second speaker's voice profile is provided by second training at the destination terminal, and wherein

said first training comprises said first speaker speaking a number of words, which can be pre-determined words, expected words, or unexpected but recognized words, and

said second training comprises said second speaker speaking a number of words, which can be pre-determined words, expected words, or unexpected but recognized words.

21. (Previously Presented) The method of claim 1, wherein the translating said speech into text comprises translating the speech into digitally coded symbols.

22. (Previously Presented) The method of claim 1, wherein the translating said speech into text comprises translating the speech into digitally coded characters.

23. (Previously Presented) The method of claim 1, further comprising:

after the ending of the communicating, determining if the telephony link is re-connected; and

continuing the communicating when the determining indicates that the telephony link is re-connected.

24. (Currently Amended) The method of claim 16, wherein the translating the first speaker's speech into the first text characters comprises translating the first speaker's speech into first digitally coded symbols representing the first text characters, and

the translating the second speaker's speech into the second text characters comprises translating the ~~first~~ second speaker's speech into second digitally coded symbols representing the second text characters.

25. (Currently Amended) The method of claim 16, wherein the translating the first speaker's speech into the first text comprises translating the first speaker's speech into first digitally coded symbols representing the first text characters, and

the translating the second speaker's speech into the second text characters comprises translating the ~~first~~ second speaker's speech into second digitally coded symbols representing the second text characters.

26. (Currently Amended) A system of communicating speech using very low digital data bandwidth, comprising:

a digital packet network;

a source terminal and a destination terminal operatively connected over the packet network on bi-directional digital telephony link;

wherein the source terminal for a forward link serves as the destination terminal

for a reverse link on the digital telephony link;

wherein the source terminal:

distinguishes between speech and a pause in the speech communication using an ITU voice activity detection module;

provides a comfort noise simulation of background noise for each distinguished pause in the speech communication in time order with the speech, translates speech into text, communicates the text and the time-ordered simulated comfort noise of each pause in speech across the bi-directional digital telephony link to the destination terminal, determines a status of the telephony link, ends the communication if the telephony link is terminated,

generates a voice profile by recognizing words spoken by a speaker for reproduction of audible speech corresponding to words spoken by the speaker having audible qualities approximating that of the speaker, and

communicates the voice profile across the telephony link to the destination terminal, and

wherein the speaker's voice profile contains the information needed to generate the reproduced speech that substantially resembles the sound of the speaker's voice; and

wherein the destination terminal translates the text into reproduced speech ~~at~~ using the speaker's voice profile and reproduces the simulated background noise in the reproduced speech in time order with each pause in the speech at the source terminal

~~received~~ at the destination terminal.

27. (Previously Presented) The method of claim 26, wherein the source terminal translates said speech into text as digitally coded symbols.

28. (Previously Presented) The method of claim 26, wherein the source terminal translates said speech into text as digitally coded characters.

29. (Previously Presented) The method of claim 26, further comprising:

wherein, after the source terminal communicates the text the source terminal determines if the telephony link is re-connected, and upon detection of re-connection of the telephony link, continues to communicate the text to the destination terminal.